ABSTRACT OF THE DISCLOSURE

A method and apparatus are disclosed for performing sampling rate conversion of an audio signal from a first sampling rate to any of a plurality of higher sampling rates using a single set of low-pass filter coefficients. Sampling rate conversion is accomplished by effectively up-sampling, low-pass filtering, and down-sampling the audio signal to generate interpolated output samples of a second digital audio signal at any of a plurality of sampling rates. The sampling rate conversion process includes storing a fixed set of filter coefficients as a plurality of phased subsets of filter coefficients, applying samples of the audio signal to the phased subsets in a rotational manner to generate filtered samples of the audio signal, and selecting and linear interpolating between certain filtered samples to generate samples of the second digital audio signal.